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Recommended Modifications to Improve CVSD Speech-Encoding Performance

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This report contains a discussion of several continuously-variable-slope delta modulator (CVSD) modifications warranting theoretical and experimental evaluation as performance-improving techniques. These candidate techniques were derived from a survey of the applicable research literature on delta modulation (ΔM) and other waveform encoding methods. The report begins with brief background information on low-data-rate speech encoding, delta modulation, and CVSD.

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20. Abstract (Continued)

Second, the proposed techniques warranting evaluation are discussed in the context of the relevant research references and their implications. Third, comparisons are made of the proposed modifications to the CVSD with the current CVSD in terms of implementation complexity, decoder output SNR, and input power dynamic range. Finally, in terms of a likelihood of increased complexity and additional systems integration chores, the following preference ordering of the proposed techniques is recommended as a guide for further evaluation: (1a) adaptive filtering of the reconstructed speech signal or (1b) optimization of the CVSD parameters and/or decisions with respect to performance measures favoring slope-overload "noise" over granular noise; (2a) coding of encoder data with non-binary codes such as ternery or correlative codes or (2b) optimization of the CVSD parameters and/or decisions with respect to performance measures incorporating past and/or future data; and (3) adaptive (asynchronous) sampling of the input speech signal.

RECOMMENDED MODIFICATIONS TO IMPROVE CVSD SPEECH-ENCODING PERFORMANCE

INTRODUCTION

Improving the continuously-variable-slope delta modulator (CVSD) speech intelligibility and quality for certain relatively-low-data-rate speech encoding applications is an important problem in the Navy and other DOD branches [1]. The ultimate objective of this study is a modification or technique for improving CVSD performance, yet requiring a low-to-moderate increase in complexity or system integration chores. This report contains a discussion of several CVSD modifications warranting theoretical and experimental evaluation as performance-improving techniques. These candidate techniques were derived from a survey of the applicable research literature on delta modulation (ΔM) and other waveform encoding techniques.

The report begins with some background information on low-data-rate speech encoding, basic delta modulation, and the CVSD. Then the proposed techniques warranting evaluation are discussed, with the relevant research references and their implications being included in this discussion. Third, comparisons of the proposed modifications to the CVSD with the current CVSD are made with respect to implementation complexity, decoder output SNR, and input power dynamic range. Finally, the conclusion contains a summary and a recommendation for a preference ordering of these proposed techniques.

BACKGROUND

The Desire for Low-Data-Rate Speech Encoding

Secure and nonsecure digital voice transmissions are increasingly required over links such as HF radio, mobile radio, and switched analog telephone networks which permit transmission rates of only 10 kb/s or less [2,3]. In keeping within current data-transmission-rate classifications, digitally encoded speech transmitted at rates of 10 kb/s or less is considered low-data-rate speech [3,4].

The Navy desires low-data-rate speech encoders for application to several important voice communication systems which operate in a variety of transmission environments. Other DOD agencies and civilian communication companies share some of the same applications. These systems and environments include:

- Satellite links and other wideband channels, which are capable of multiplexing several narrowband channels and are better used by low-data-rate encoders [5,6];

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- Transmission media subjected to restrictive spectrum allocation and conservation specifications;
- RF transmission links having fixed and limited available power, which can achieve superior SNR at the receiver by using low data rate;
- Covert communication situations in which the principal approach is to transfer the minimum required information at the minimum practical data rate [7]; and
- Underwater acoustic communication channels having characteristics which admit only low transmission rates.

Only two widely known techniques will provide the required low data rates: spectral deconvolution techniques (e.g., vocoders and other analysis/synthesis methods [3,5,6,8, 9] and ΔM techniques [3].

ΔM Techniques

The basic ΔM encoder (Fig. 1) consists of an adder, a hard limiter (a comparator with binary output), a sampler, and an integrator. All of these components are simply and cheaply implementable. The ΔM decoder at the receiver is just the integrator of the ΔM encoder feedback loop, followed by a low-pass filter.

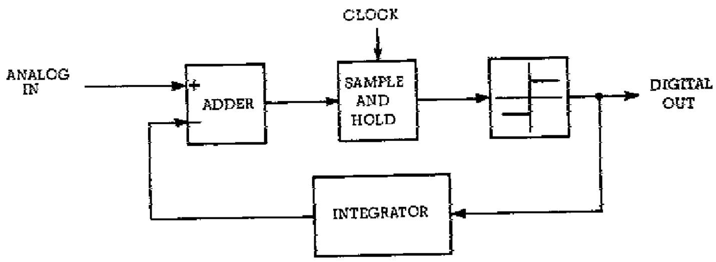

Fig. 1 — Basic delta modulator

The ΔM can be viewed as a one-bit differential-pulse-code-modulation (DPCM) system with feedback in which good reconstruction performance at one bit per sample depends on a highly correlated input signal with respect to the sampling rate. For sampling rates above 32 kb/s the correlation of speech samples is high enough for the basic ΔM and/or the current adaptive ΔM to perform with acceptable intelligibility and quality. Adaptive ΔM increases the accuracy (generally in a mean-squared-error sense) of the reconstructed signal by varying the output data pulse heights (effectively the encoder step size) at the integrator input. This variation is usually based on the input signal's slope or average level [3-5,8,10-26]. This added degree of freedom for adaptive ΔM yields intelligible speech at 16 kb/s even through noisy channels.

The Current Navy ΔM: CVSD

The CVSD is an adaptive ΔM currently under consideration for use as a speech waveform digitizer by the Navy and several other DOD agencies [14]. In the CVSD the slope-overload level of the encoder is varied at a syllabic rate. Consequently, subjective performance degrading effects, associated with instantaneous companding or transmission errors, are further reduced, and the input speech dynamic range is improved.

The slope control signal for adjusting the slope-overload level is derived directly from the three most recent encoder output bits, thereby allowing the ΔM output pulse height (step size) compression and expansion, performed at the encoder and decoder, to track very well without the necessity of additional speech-signal-envelope information. A slope-overload condition is assumed if the three most recent output bits are alike. This occurs when the input speech signal is increasing or decreasing so fast that the reconstructed signals cannot keep up with it. In this situation the CVSD encoder (the step-size algorithm) temporarily increases the step size, so that the reconstructed signal more closely tracks the original signal. During nonoverload conditions the step-size decays exponentially to a level which minimizes granular noise. The decay time-constant is comparable to a syllabic time interval. The maximum and minimum step-size levels are chosen for subjectively good all-round performance.

CVSD MODIFICATIONS WARRANTING EVALUATION

Optimizing Parameters to Other Performance Measures

The standard performance measure used for ΔM designs is the signal-to-quantization-noise power ratio (SNR), with quantization noise including the slope overload distortion ("noise") in addition to the granular noise. As usual, the popularity of SNR in this context is due to its measurability and mathematical tractability. However the response of the human ear is not matched to such an objective performance measure as the SNR; consequently many researchers have sought more subjective performance measures, often with little success. Nevertheless some encouraging results have been reported and warrant further evaluation in the desired context. Proposed in particular is evaluation of performance measures incorporating the effects of past, present and future decisions and performance measures favoring slope-overload noise over granular noise.

Performance Measures Incorporating the Effects of Past, Present, and Future Decisions

Acknowledgment of past, present, and future effects generated by each coding decision admits the influence of past and/or future decisions and effects on the current decision. As a result, more control is gained over the accumulative effects of each decision. Zetterberg and Uddenfeldt [4] consider a delayed decision approach in which the current decision on the step size and the ΔM output at time step n is based on the m delayed input samples $\{x_n, ..., x_{n+m-1}\}$. Two performance measures are employed. They are weighted sums of the mean-squared reconstruction errors incurred over those m delayed samples. The performance measure, weighting more heavily against the error (quantization noise power) within the speech band, gives better SNR performance, since decoder output filtering is then more effective.

Song [21] and Song et al. [22] use approximate minimum mean-squared-error simulation (at the decoder) and one-step-ahead prediction (in the encoder feedback loop) algorithms to reconstruct a Gaussian-Markov input signal. The current predictor value is conditioned on the last two estimator values and the last two ΔM output values. Simulations of their method show a SNR quite insensitive to the input signal power level. The SNR level is also comparable to other ΔM configurations which are operating at their optimum signal power level.

Jayant [18] instantaneously adapts the step size on the basis of a comparison between the two latest ΔM output digits. If the two digits are the same, the step size is changed by the factor $P > 1$; otherwise the factor is $-Q$, where $P \cdot Q = 1$.

Performance Measures Favoring Slope-Overload Noise Over Granular Noise

Several researchers have investigated the perception of slope-overload noise and the subjective preference of slope-overload noise over granular noise. The results of Jayant and Rosenberg [27] indicate that speech samples exhibiting the minimum degradation on an objective quantization-noise-power basis are not subjectively the most preferred samples. They also show that a subjectively prefered ΔM generates *greater* slope-overload and lesser granularity than the objectively optimum ΔM. As a possible explanation Jayant and Rosenberg suggest that granularity is explicitly perceivable as background noise, while slope-overload "noise" exists only in relation to an original signal which is not known to the listener. They feel that this preference for greater slope-overload and lesser granularity may be more significant at data rates lower than those considered.

Levitt et al. [28] investigated the perception of speech distortion resulting from slope overload. Their research demonstrates the inappropriateness of the SNR as a viable performance measure, since there can be a substantial variation of the SNR at the level of just-perceptible distortion. They also recommend a more appropriate parameter for slope-overload distortion which is a function of the truncated portion of the input signal's time derivative beyond the maximum absolute slope level of the ΔM.

Greefkes [17] used these results in designing a ΔM system for which intelligible speech at 7.2 kb/s is claimed. His system attempts to maintain the ΔM more often in a slope-overload condition rather than allow granular noise to occur.

Applying Nonbinary and Nonsynchronous Signaling Techniques

It is well known that speech signals are hightly redundant and contain a large percentage of quiet periods. Eger and Campanella [29] claim that 60% of conversational speech is quiet. The standard binary synchronous output waveform is not particularly suited to coding a speech waveform. For example, the only means of representing constant-speech-level intervals is by an alternating synchronous waveform, which yields a substantial granular noise component. In addition the CVSD, like other adaptive ΔM systems, suffers rapid performance degradation in noisy-channel environments and thereby causes this otherwise attractive speech digitization technique to be unsatisfactory for some important applications. As approaches to solving the problem of granular noise, techniques of nonbinary waveform coding and amplitude sampling more closely matched to the input waveform characteristics are proposed; and as approaches to solving the problem of noisy channels, coding techniques which add redundancy without substantially increasing the desired low signaling rate are proposed.

Waveform Coding and Amplitude Sampling More Closely Matched to the Input Waveform Characteristics

Speech redundancy can be further exploited by replacing the binary digit code with a ternery digit code or by applying adaptive sampling techniques. Inose et al. [30] theoretically and experimentally demonstrate that a ternery digit ΔM output code with symbols -1, 0, and 1 used the symbols -1 and 1 at less than 2/5 the rate of a binary digit code (-1, 1) for a white-noise signal and equivalent output SNR. It follows that further reduction of ± 1 symbol rates are expected when speech signals are considered, since speech consists of a high percentage of fairly-constant-level periods. The immediate result is a reduction in granular noise at the expense of an additional decision level. Halijak and Tripp [31] suggest a technique which may give similar results; they recommend that the ΔM comparator have a small dead zone to reduce granular noise.

Adaptive sampling (asynchronous sampling or, equivalently, aperiodic sample times) of the speech signal is another technique that has been investigated for better matching of the encoding system to the speech waveform characteristics [30,32,33]. These techniques are based on taking a sample whenever the speech signal or its derivative reaches a certain threshold value. The ternery digit system of Inose et al. [30] has some performance improvement whenever the ± 1 symbols from the (-1, 0, 1) code are transmitted at the instant of some amplitude threshold crossings. Hawkes and Simonpieri [32] describe an asynchronous ΔM system with memory. Their system performs below that of the "ideal" asynchronous system [30] but does not require timing information transmitted to the receiver. As an example, if the last two ΔM output digits have the same sign, then the input is assumed to be growing in absolute value, and the sampling interval should be reduced. Conversely, if the last two output digits do not have the same sign, then the input is assumed to be relatively constant, and the sampling interval should be increased. Since the adaptation algorithm depends on only the ΔM output, the decoder does not need additional information to track the adaptive sampling times of the encoder. Mostafa and El-Hagry [33] proposed an adaptive sampling technique for which the sampling times depend on the speech-signal time derivative. The effect here is an increase in the sampling rate during that portion of the speech waveform having the larger, higher frequency components or, equivalently, an increase in the sampling rate when the speech waveform has large and rapid changes.

Channel Coding Techniques to Improve Noisy Transmission Performance

The traditional solution to the problem of noisy channels is to apply the standard channel coding techniques which add redundancy digits to the output code. However for the desired applications the resulting increase in signaling rate is undesirable. Therefore a technique for adding redundancy may be beneficial if it does not substantially increase the signal rate. In other words a desirable technique would transform the output code into a code with the same signaling rate and with known correlative properties that may allow error detection and correction.

Lender [34-36], Sekey [37], and Wolf [38] have investigated some of these correlation-introducing techniques. Lender and Sekey presented a technique known as the polybinary technique, which transforms a binary digit sequence into a n-ary digit sequence ($n > 2$) with the same signaling rate. The transformation creates correlation within the n-ary digit sequence in a specified manner, thus allowing error detection. The case of $n = 3$, known as the duobinary technique, appears to be the most developed [34-37]. This property of the polybinary technique is gained however at the expense of a decrease in threshold-level widths, since $n - 1$ threshold levels are required. Nevertheless it is uncertain whether the duobinary (polybinary) technique with some form of error correction (such as assigning a likely speech waveform state of value based on context when an error is detected) performs better than the standard binary digit code for nonerrorfree transmission requirements.

Wolf investigated a more general transformation of binary sequences which included the polybinary technique. Some form of error correction is implied when he discusses the need for decoding decisions based on information-data redundancy and on code-transformation knowledge.

Adaptive Filtering of Reconstructed Speech Signals

Eger and Campanella [9] have proposed adaptively filtering a speech signal to make smaller the average bandwidth required in adaptive-channel-bandwidth multiplexed systems. They exploit the redundancy and quiet periods of speech, which are characterized by an average conversational speech signal that is 60% quiet, with 10.4% of the signal confined to a 1 kHz bandwidth, 13.2% to a 2-kHz bandwidth, and 16.4% to a 4 kHz bandwidth. Likewise, adaptive filtering may also be used to reduce the subjectively objectionable granular noise in the ΔM reconstructed speech signal. In this case an adaptive-filter bandwidth would better match the short-term conditions on a subjective and an objective basis.

If the filter adaptation algorithm is based solely on the ΔM encoder output data stream, as is the step-size algorithm, then additional information need not be transmitted to the ΔM decoder. For example, the adaptation may be in choosing one out of four filters having bandwidths of 0 kHz, 1 kHz, 2 kHz, and 4 kHz. Goodman [15] and Goodman and Greenstein [39] have already shown all-digital implementations of delta modulators. Consequently filter adaptation may be controlled by all-digital methods.

COMPARISON OF MODIFICATIONS WITH CVSD

Implementation Complexity

Asynchronous data rates and adaptive sampling techniques require asynchronous transmission, encryption, decryption, demodulation, and reconstruction when buffering and data time-labeling methods are not applied. Consequently the proposed techniques, if acceptable under TRANSEC (transmission security) considerations because of possible intelligence unveiling, may not be readily acceptable from a systems integration viewpoint (buffering and its consequences may be preferred) unless their performance improvement justifies the interfacing equipment modification. The increase in complexity at the encoder and decoder is expected to be small-to-moderate for asynchronous encoding and decoding alone. Most of this increase will probably be at the encoder.

The increase in encoder and decoder complexity for synchronous ternery digit coding is expected to be about the same level or less compared with the asynchronous binary digit case. However the systems integration task may be acceptable and certainly easier if the ternery symbols are mapped into quartenary symbols, thereby allowing more easily implementable four-level synchronous transmission, encryption, decrytion, and demodulation. Asynchronous ternery digit coding may require twice the complexity increase required by the synchronous method.

Real-time adaptive algorithms based on other than instantaneous mean-squared-error performance measures, possibly incorporating past and/or future data, require storage and computing capability at the encoder and decoder. This task may be easily accomplished by a basic microprocessor or by some simple arrangement of standard logic blocks.

Adaptive-filter implementation at the decoder would probably require a low-to-moderate increase in complexity. The adaptive filter may be implemented using analog or digital techniques, although some combination would be more likely. The adaptive filter may be a basic analog filter with digitally alterable parameters or components and thereby capable of realizing N desired filters. The adaptive-filter control signal may be derivable from the received ΔM output data stream. If so, the adaptive-filter control algorithm may be adequately realized by all-digital techniques, since only one out of N filter configurations is chosen at each decision stage.

SNR Performance

Although it has been stated here and elsewhere that the subjective performance of a ΔM system is not directly related to the decoder output SNR, the obvious indirect relationship (e.g., a large SNR yields intelligible and good-quality speech), as well as other attributes, make the SNR a universal performance measure. Hence some SNR performance comparisons of these proposed techniques with CVSD are in order.

The SNR should be improved for adaptive-sampling, adaptive-filtering, and ternery-digit-coding techniques. The granular noise would be considerably reduced for ternery digit coding or adaptive filtering, and the slope-overload noise would be reduced for adaptive sampling. Better SNR would also be expected when ΔM parameter optimization is for performance measures weighting against a reconstruction error function incorporating past or future data. However the SNR may not be better for performance measures weighting against none or only part of the reconstruction error; for example, weighting against only the granular noise may improve the subjective performance while decreasing the SNR by increasing the slope-overload noise.

Input Dynamic Range

The criterion for selecting adaptive sampling times and for selecting the no-change levels (the 0 symbol) in ternery digit coding determines the low-level-input-signal sensitivity of ΔM using those techniques. However both techniques give a no-signal indication (101010... for adaptive sampling and 000000... for ternery coding) when the lowest step-size threshold is not exceeded. Consequently the low-level-input-signal SNR performance for these techniques should be better than CVSD performance for the same lowest step-size threshold just by virtue of the additional degree of freedom.

The input dynamic range performance (SNR vs input signal level) falls off at high input-signal levels, depending on the largest step size and the adaptive step-size algorithm. For equivalent step sizes and algorithms the adaptive sampling techniques should show a more gradual fall off, since another degree of freedom is allowed for SNR optimization (fewer constraints are imposed). Likewise, the ternery digit coding technique should show improved performance at high input-signal levels, since granular noise is reduced.

The adaptive-filtering technique should show improved dynamic range performance over CVSD, since reconstruction "noise" (error) outside the current adaptive-filter bandwidth is substantially reduced. As a result the average "noise" power is reduced. Finally, those techniques using performance measures (incorporating reconstruction errors averaged over some past and/or future data) should also have better input-dynamic-range performance compared with the CVSD. This follows because, for equivalent step sizes, more data are available at each decision point.

SUMMARY AND CONCLUSIONS

In this report several techniques or modifications were proposed for improving CVSD performance at low data rates. Low data rates were defined here as less than 10 kb/s. A survey of available research literature on delta-modulation and speech-encoding techniques was the basis for this report. The proposed modifications to the CVSD are characterized as follows:

- Adaptive (asynchronous) sampling of the input speech signal;
- Adaptive filtering of the reconstructed speech signal;
- Optimization of CVSD parameters and/or decisions to performance measures incorporating past and/or future data;
- Optimization of CVSD parameters and/or decisions to performance measures favoring slope-overload "noise" over granular noise; and
- Coding of encoder data with nonbinary codes such as ternery or correlative (e.g., polybinary) codes.

These techniques were further explained, and the justification for their consideration was discussed. For comparison, some assessment of their impact on CVSD complexity, input dynamic range, and SNR was also made.

In terms of a likelihood of increased complexity and additional system integration chores, we recommend the following preference ordering as a guide for further evaluation of the proposed techniques:

1a. Adaptive filtering of the reconstructed speech signal or

1b. Optimization of the CVSD parameters and/or decisions with respect to performance measures favoring slope-overload "noise" over granular noise;

2a. Coding of encoder data with nonbinary codes such as ternery or correlative codes or

2b. Optimization of the CVSD parameters and/or decisions with respect to performance measures incorporating past and/or future data; and

3. Adaptive (asynchronous) sampling of the input speech signal.

Therefore, as a goal, this report has attempted to indicate that further theoretical and experimental evaluations of these techniques for improving CVSD performance are warranted based on probable practical acceptability and on the implications of pertinent published research results.

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